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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 10/081,857	Applicant(s) FERNANDEZ-CORBATON ET AL.	
	Examiner Vineeta S. Panwalkar	Art Unit 2631	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *20 February 2002*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-48* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-48* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *20 February 2002* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Oath/Declaration

1. The Combined Declaration/ Power of Attorney is objected to because it is not signed and dated by the applicants. A letter dated 03/19/2002, informing the applicants about the unsigned declaration, was mailed by the Patent Office.

Double Patenting

2. A rejection based on double patenting of the "same invention" type finds its support in the language of 35 U.S.C. 101 which states that "whoever invents or discovers any new and useful process ... may obtain a patent therefor ..." (Emphasis added). Thus, the term "same invention," in this context, means an invention drawn to identical subject matter. See *Miller v. Eagle Mfg. Co.*, 151 U.S. 186 (1894); *In re Ockert*, 245 F.2d 467, 114 USPQ 330 (CCPA 1957); and *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970).

A statutory type (35 U.S.C. 101) double patenting rejection can be overcome by canceling or amending the conflicting claims so they are no longer coextensive in scope. The filing of a terminal disclaimer cannot overcome a double patenting rejection based upon 35 U.S.C. 101.

2a. Claims 1-5,7-17,19-28,30-36,38-45, 47 and 48 are provisionally rejected under 35 U.S.C. 101 as claiming the same invention as that of claims 1-5,7-17,19-27,29-35,37-44, 46 and 47 of copending Application No. 10/115,210. This is a provisional double patenting rejection since the conflicting claims have not in fact been patented.

Claims of App # 10/081,857: (Current application)	**Claims of App # 10/115,210:** (Copending application)
1. A method of filtering a plurality of samples, comprising: adapting a plurality of filter coefficients; and filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each remaining filter coefficient to one of the samples, and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.	1. A method of filtering a plurality of samples, comprising: adapting a plurality of filter coefficients; and filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each remaining filter coefficient to one of the samples, and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.
2. The method of claim 1 wherein the filtering of samples comprises multiplying one of the filter coefficients with said parameter, multiplying each of the remaining filter coefficients with its respective sample, and summing the parameter and the samples.	2. The method of claim 1 wherein the filtering of samples comprises multiplying one of the filter coefficients with said parameter, multiplying each of the remaining filter coefficients with its respective sample, and summing the parameter and the samples.
3. The method of claim 1 wherein the adaptation of the filter coefficients comprises using a least square algorithm.	3. The method of claim 1 wherein the adaptation of the filter coefficients comprises using a least square algorithm.
4. The method of claim 3 wherein the least square algorithm comprises a least mean square (LMS) algorithm.	4. The method of claim 3 wherein the least square algorithm comprises a least mean square (LMS) algorithm.
5. The method of claim 1 wherein the parameter comprises a fixed value.	5. The method of claim 1 wherein the parameter comprises a fixed value.
7. The method of claim 1 further comprising monitoring a DC bias of the samples, and reducing the DC bias if it exceeds a threshold.	7. The method of claim 1 further comprising monitoring a DC bias of the samples, and reducing the DC bias if it exceeds a threshold.

8. The method of claim 1 further comprising notch filtering the samples.	8. The method of claim 1 further comprising notch filtering the samples.
9. The method of claim 8 wherein the notch is substantially at DC.	9. The method of claim 8 wherein the notch is substantially at DC.
10. The method of claim 1 wherein the adaptation of the filter coefficients is further a function of a plurality of locally generated samples.	10. The method of claim 1 wherein the adaptation of the filter coefficients is further a function of a plurality of locally generated samples.
11. The method of claim 10 wherein the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.	11. The method of claim 10 wherein the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.
12. A receiver, comprising: an analog-to-digital converter configured to sample an analog signal to produce a plurality of samples; and a filter having a coefficient generator configured to adapt a plurality of filter coefficients, the filter being configured to apply one of the filter coefficients to a parameter, apply each of the remaining filter coefficients to one of the samples, and combine the parameter and the samples, the adaptation of the filter coefficients being a function of the combined parameter and samples.	12. A receiver, comprising: an analog-to-digital converter configured to sample an analog signal to produce a plurality of samples; and a filter having a coefficient generator configured to adapt a plurality of filter coefficients, the filter being configured to apply one of the filter coefficients to a parameter, apply each of the remaining filter coefficients to one of the samples, and combine the parameter and the samples, the adaptation of the filter coefficients being a function of the combined parameter and samples.
13. The receiver of claim 12 wherein the filter further comprises a first multiplier configured to multiply said one of the filter coefficients with the parameter, a	13. The receiver of claim 12 wherein the filter further comprises a first multiplier configured to multiply said one of the filter coefficients with the parameter, a

second multiplier configured to multiply each of the remaining filter coefficients with its respective sample, and an adder configured to sum the parameter and the samples.	second multiplier configured to multiply each of the remaining filter coefficients with its respective sample, and an adder configured to sum the parameter and the samples.
14. The receiver of claim 13 wherein the filter further comprises a delay element configured to serially receive the samples from the analog-to- digital converter, and wherein the second multiplier is further configured to multiply each of the remaining filter coefficients with its respective sample from the delay element.	14. The receiver of claim 13 wherein the filter further comprises a delay element configured to serially receive the samples from the analog-to- digital converter, and wherein the second multiplier is further configured to multiply each of the remaining filter coefficients with its respective sample from the delay element.
15. The receiver of claim 12 wherein the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm.	15. The receiver of claim 12 wherein the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm.
16. The receiver of claim 15 wherein the least square algorithm comprises a least mean square (LMS) algorithm.	16. The receiver of claim 15 wherein the least square algorithm comprises a least mean square (LMS) algorithm.
17. The receiver of claim 12 wherein the parameter comprises a fixed value.	17. The receiver of claim 12 wherein the parameter comprises a fixed value.
19. The receiver of claim 12 further comprising an outer correction loop configured to monitoring a DC bias of the samples generated by the analog-to-digital converter, and reducing the DC bias if it exceeds a threshold.	19. The receiver of claim 12 further comprising an outer correction loop configured to monitoring a DC bias of the samples generated by the analog-to-digital converter, and reducing the DC bias if it exceeds a threshold.
20. The receiver of claim 12 further comprising a notch filter configured to filter the samples.	20. The receiver of claim 12 further comprising a notch filter configured to filter the samples.

21. The receiver of claim 12 further comprising a notch filter configured to filter the samples. 22. The receiver of claim 21 wherein the notch filler is further configured with a notch substantially at DC.	21. The receiver of claim 20 wherein the notch filler is further configured with a notch substantially at DC.
23. The receiver of claim 12 wherein the receiver further comprises a sample generator configured to generate a plurality of locally generated samples, and wherein the coefficient generator is further configured to adapt the filter coefficient as a function of the locally generated samples.	22. The receiver of claim 12 wherein the receiver further comprises a sample generator configured to generate a plurality of locally generated samples, and wherein the coefficient generator is further configured to adapt the filter coefficient as a function of the locally generated samples.
24. The method of claim 23 wherein the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.	23. The method of claim 22 wherein the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.
25. A filter, comprising: a delay element configured to serially receive a plurality of samples; a coefficient generator configured to adapt a plurality of coefficients; a first multiplier configured to multiply said one of the filter coefficients with the parameter; a second multiplier configured to multiply each remaining filter coefficient with one of the samples from the delay element; and an adder configured to sum the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the summed parameter and	24. A filter, comprising: a delay element configured to serially receive a plurality of samples; a coefficient generator configured to adapt a plurality of coefficients; a first multiplier configured to multiply said one of the filter coefficients with the parameter; a second multiplier configured to multiply each remaining filter coefficient with one of the samples from the delay element; and an adder configured to sum the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the summed parameter and

samples.	samples.
26. The filter of claim 25 wherein the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm.	25. The filter of claim 24 wherein the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm.
27. The filter of claim 26 wherein the least square algorithm comprises a least mean square (LMS) algorithm.	26. The filter of claim 25 wherein the least square algorithm comprises a least mean square (LMS) algorithm.
28. The filter of claim 25 wherein the parameter comprises a fixed value.	27. The filter of claim 24 wherein the parameter comprises a fixed value.
30. The filter of claim 25 wherein the coefficient generator is further configured to receive a plurality of locally generated samples, and adapt the filter coefficient as a function of the locally generated samples.	29. The filter of claim 24 wherein the coefficient generator is further configured to receive a plurality of locally generated samples, and adapt the filter coefficient as a function of the locally generated samples.
31. The filter of claim 30 wherein the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.	30. The filter of claim 29 wherein the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.
32. Computer-readable media embodying a program of instructions executable by a computer program to perform a method of adapting filter coefficients, the method comprising: adapting a plurality of filter coefficients; and filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each remaining filter	31. Computer-readable media embodying a program of instructions executable by a computer program to perform a method of adapting filter coefficients, the method comprising: adapting a plurality of filter coefficients; and filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each remaining filter

coefficient to one of the samples, and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.	coefficient to one of the samples, and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.
33. The computer-readable media of claim 32 wherein the filtering of samples multiplying said one of the filter coefficients with the parameter, multiplying each of the remaining filter coefficients with its respective sample, and summing the parameter and the samples.	32 The computer-readable media of claim 31 wherein the filtering of samples multiplying said one of the filter coefficients with the parameter, multiplying each of the remaining filter coefficients with its respective sample, and summing the parameter and the samples.
34. The computer-readable media of claim 32 wherein the adaptation of the filter coefficients comprising using a least square algorithm.	33. The computer-readable media of claim 31 wherein the adaptation of the filter coefficients comprising using a least square algorithm.
35. The computer-readable media of claim 34 wherein the least square algorithm comprises a least mean square (LMS) algorithm.	34. The computer-readable media of claim 33 wherein the least square algorithm comprises a least mean square (LMS) algorithm.
36. The computer-readable media of claim 32 Wherein the parameter comprises a fixed value.	35. The computer-readable media of claim 31 Wherein the parameter comprises a fixed value.
38. The computer-readable media of claim 32 wherein the adaptation of the filter coefficients is further a function of a plurality of locally generated samples.	37. The computer-readable media of claim 31 wherein the adaptation of the filter coefficients is further a function of a plurality of locally generated samples.
39. The computer-readable media of claim 38 wherein the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.	38. The computer-readable media of claim 37 wherein the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.

40. A filter, comprising: means for adapting a plurality of filter coefficients; and means for filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each of the remaining filter coefficients to one of the samples and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.	39. A filter, comprising: means for adapting a plurality of filter coefficients; and means for filtering a plurality of samples by applying one of the filter coefficients to a parameter, applying each of the remaining filter coefficients to one of the samples and combining the parameter and the samples; wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.
41 . The filter of claim 40 wherein the means for filtering the samples comprises means for multiplying said one of the filter coefficients with the parameter, means for multiplying each of the remaining filter coefficients with its respective sample, and means for summing the parameter and the samples.	40. The filter of claim 39 wherein the means for filtering the samples comprises means for multiplying said one of the filter coefficients with the parameter, means for multiplying each of the remaining filter coefficients with its respective sample, and means for summing the parameter and the samples.
42. The filter of claim 41 wherein the means for filtering the samples further comprises means for serially receiving the samples.	41. The filter of claim 40 wherein the means for filtering the samples further comprises means for serially receiving the samples.
43. The filter of claim 40 wherein the means for adapting the filter coefficients uses a least square algorithm.	42. The filter of claim 39 wherein the means for adapting the filter coefficients uses a least square algorithm.
44. The filter of claim 43 wherein the least square algorithm comprises a least mean square (LMS) algorithm.	43. The filter of claim 42 wherein the least square algorithm comprises a least mean square (LMS) algorithm.
45. The filter of claim 40 wherein the parameter comprises a fixed value.	44. The filter of claim 39 wherein the parameter comprises a fixed value.
47. The filter of claim 40 wherein the adaptation of the filter	46. The filter of claim 39 wherein the adaptation of the filter

coefficients are further a function of the locally generated samples.	coefficients are further a function of the locally generated samples.
48. The filter of claim 47 wherein the adaptation of the filter coefficients are performed by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.	47. The filter of claim 46 wherein the adaptation of the filter coefficients are performed by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples.

3. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. See *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970);and, *In re Thorington,* 418 F.2d 528, 163 USPQ 644 (CCPA 1969).
A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent is shown to be commonly owned with this application. See 37 CFR 1.130(b).
Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

3a. Claims 6,18,29,37 and 46 are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 6,18,28,36 and 45 of copending Application No. 10/115,210. Although the conflicting claims are not identical, they are not patentably distinct from each other because the disclosures of both applications do not show any criticality in choosing the value of the claimed parameter.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

Claims of App # 10/081,857: (Current application)	**Claims of App # 10/115,210:** (Copending application)
6. The method of claim 5 wherein the samples have an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples.	6. The method of claim 5 wherein the samples have an average power value, and wherein the fixed value of the parameter is substantially equal to the square root of the average power value of the samples.
18. The receiver of claim 17 wherein the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples.	18. The receiver of claim 17 wherein the samples comprise an average power value, and wherein the fixed value of the parameter is substantially equal to the square root of the average power value of the samples.
29. The filter of claim 28 wherein the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples.	28. The filter of claim 27 wherein the samples comprise an average power value, and wherein the fixed value of the parameter is substantially equal to the square root of the average power value of the samples.
37. The computer-readable media of claim 36 wherein the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples.	36. The computer-readable media of claim 35 wherein the samples comprise an average power value, and wherein the fixed value of the parameter is substantially equal to the square root of the average power value of the samples.
46. The filter of claim 45 wherein the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the	45. The filter of claim 44 wherein the samples comprise an average power value, and wherein the fixed value of the parameter is substantially equal to the square

samples.	root of the average power value of the samples.

Claim Rejections - 35 USC § 112

The following is a quotation of the second paragraph of 35 U.S.C. 112:

The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

4. Claim 25 recites the limitation "the parameter" in line 8 of the claim. There is insufficient antecedent basis for this limitation in the claim.

Claim Rejections - 35 USC § 102

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

5. Claims 1-6,10-18,23-31and 40-48 are rejected under 35 U.S.C. 102(b) as being anticipated by Yedid et al. (US 5526377), hereafter referred to as Yedid.

5a. Regarding claim 1, Yedid discloses a transversal filter for minimizing non-linear distortion in signals comprising:

- A method of filtering a plurality of samples, comprising:

adapting a plurality of filter coefficients(coefficients c_i in Fig. 6); and filtering a plurality of samples by applying one of the filter coefficients to a parameter (provided by DC tap 116 in Fig. 6), applying each remaining filter coefficient to one of the samples, and combining the parameter and the samples (Summer 115 in Fig. 6 combines the samples and parameter); wherein the adaptation of the filter coefficients is a function of the combined parameter and samples (Fig. 6 and column 8, lines 31-60).

5b. Regarding claim 2, Yedid discloses a method wherein:

- the filtering of samples comprises multiplying one of the filter coefficients with said parameter (the DC tap value is multiplies by a coefficient at multiplier 114 in Fig. 6), multiplying each of the remaining filter coefficients with its respective sample (inputs to summers 111-1 through 111-4 are the products of filter coefficients c_i and their respective samples), and summing the parameter and the samples (Summer 115 in Fig. 6 combines the samples and parameter). (Fig. 6 and column 8, lines 31-60).

5c. Regarding claim 3, Yedid discloses a method wherein:

- the adaptation of the filter coefficients comprises using a least square algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein

equation (1) is the update equation used in Least Mean Square (LMS) algorithm. LMS is a least square algorithm).

5d. Regarding claim 4, Yedid also discloses a method wherein:

- the least square algorithm comprises a least mean square (LMS) algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm).

5e. Regarding claim 5, Yedid further discloses a method wherein:

- the parameter comprises a fixed value (Fig. 6 and column 8, lines 31-60, wherein the fixed reference value implies that the parameter has a fixed value).

5f. Regarding claim 6, Yedid further discloses a method wherein:

- the samples have an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples (Column 4, lines 37-58. The symbol values imply that the samples have an average value. The DC reference value i.e. the claimed parameter is employed as a pseudo-symbol, implying that the fixed value of the parameter is substantially equal to the average value of the symbols i.e. the claimed samples).

5g. Regarding claim 10, Yedid also discloses a method wherein:

- the adaptation of the filter coefficients is further a function of a plurality of locally generated samples (samples in delay line 101 of Fig. 6 are generated locally due to an analog to digital conversion by an analog to digital converter (ADC) such as block 33 of Fig. 1. In Fig.1, the ADC is followed by an echo canceller which uses the adaptive filter. In column 6, lines 44-53, equation 1 explains how the coefficients are adapted as a function of the delay line samples 101.).

5h. Regarding claim 11, Yedid further discloses a method wherein:

- the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples (Column 2, lines 9-22, where the LMS algorithm is a minimum mean square error algorithm and Column 6, lines 43-53, wherein equation (1) is the update equation used in LMS algorithm).

5i. Regarding claim 12, Yedid discloses a transversal filter for minimizing non-linear distortion in signals wherein is disclosed:

- A receiver comprising:
- An analog-to-digital converter configured to sample an analog signal to produce a plurality of samples (Block 33 of Fig. 1); and

- a filter having a coefficient generator (Block 107 in Fig. 6) configured to adapting a plurality of filter coefficients(coefficients c_i in Fig. 6); the filter being configured to apply one of the filter coefficients to a parameter (provided by DC tap 116 in Fig. 6), apply each of the remaining filter coefficients to one of the samples, and combine the parameter and the samples (Summer 115 in Fig. 6 combines the samples and parameter) , the adaptation of the filter coefficients being a function of the combined parameter and samples. (Fig. 1, Fig. 6 and column 8, lines 31-60).

5j. Regarding claim 13, Yedid further discloses an apparatus wherein:

- the filter further comprises a first multiplier configured to multiply said one of the filter coefficients with the parameter (Block 114 in Fig. 6), a second multiplier configured to multiply each of the remaining filter coefficients with its respective sample (the samples from black 101 in Fig.6 are multiplied by their respective updated coefficients c_i in Fig. 6), and an adder configured to sum the parameter and the samples (Block 115 in Fig. 6). (Fig. 1, Fig. 6 and column 8, lines 31-60).

5j. Regarding claim 14, Yedid further discloses an apparatus wherein:

- the filter further comprises a delay element configured to serially receive the samples (Block 101 in Fig. 6) from the analog-to- digital

converter, and wherein the second multiplier is further configured to multiply each of the remaining filter coefficients with its respective sample from the delay element(the samples from black 101 in Fig.6 are multiplied by their respective updated coefficients c_i in Fig. 6). (Fig. 1, Fig. 6 and column 8, lines 31-60).

5k. Regarding claim 15, Yedid further discloses apparatus wherein:

- the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm. LMS is a least square algorithm).

5l. Regarding claim 16, Yedid further discloses apparatus wherein:

- The receiver of claim 15 wherein the least square algorithm comprises a least mean square (LMS) algorithm. (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm).

5m. Regarding claim 17, Yedid also discloses apparatus wherein:

- the parameter comprises a fixed value. (Fig. 6 and column 8, lines 31 60, wherein the fixed reference value implies that the parameter has a fixed value).

5n. Regarding claim 18, Yedid also discloses apparatus wherein:

- the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples. (Column 4, lines 37-58. The symbol values imply that the samples have an average value. The DC reference value i.e. the claimed parameter is employed as a pseudo-symbol, implying that the fixed value of the parameter is substantially equal to the average value of the symbols i.e. the claimed samples).

5o. Regarding claim 23, Yedid discloses apparatus further comprising:

- a sample generator (block 33 of Fig. 1) configured to generate a plurality of locally generated samples, and wherein the coefficient generator is further configured to adapt the filter coefficient as a function of the locally generated samples (samples in delay line 101 of Fig. 6 are generated locally due to an analog to digital conversion by an analog to digital converter (ADC) such as block 33 of Fig. 1. In Fig.1, the ADC is followed by an echo canceller which uses the adaptive filter. In column

6, lines 44-53, equation 1 explains how the coefficients are adapted as a function of the delay line samples 101).

5p. Regarding claim 24, Yedid also discloses apparatus and method wherein:

- the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples. (Column 2, lines 9-22, where the LMS algorithm is a minimum mean square error algorithm and Column 6, lines 43-53, wherein equation (1) is the update equation used in LMS algorithm).

5q. Regarding claim 25, rejection is based upon the assumption that the first occurrence of “the parameter” in claim 25 is actually “a parameter”. Thus, Yedid discloses a transversal filter for minimizing non-linear distortion in signals comprising:

- a delay element configured to serially receive a plurality of samples (Block 101 of Fig. 6); a coefficient generator (Block107 of Fig. 6) configured to adapt a plurality of coefficients (coefficients c_i in Fig. 6); a first multiplier (Block 114 in Fig. 6) configured to multiply said one of the filter coefficients with a parameter(provided by DC tap 116 in Fig. 6); a second multiplier configured to multiply each remaining filter coefficient with one of the samples from the delay element(the samples from black

101 in Fig.6 are multiplied by their respective updated coefficients c_i in Fig. 6); and an adder configured to sum the parameter and the samples (Summer 115 in Fig. 6 combines the samples and parameter); wherein the adaptation of the filter coefficients is a function of the summed parameter and samples. (Fig. 6 and column 8, lines 31-60).

5r. Regarding claim 26, Yedid also discloses apparatus wherein:

- the coefficient generator is further configured to adapt the filter coefficients using a least square algorithm. (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm. LMS is a least square algorithm).

5s. Regarding claim 27, Yedid also discloses apparatus wherein:

- the least square algorithm comprises a least mean square (LMS) algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm).

5t. Regarding claim 28, Yedid also discloses apparatus wherein:

- the parameter comprises a fixed value (Fig. 6 and column 8, lines 31 60, wherein the fixed reference value implies that the parameter has a fixed value).

5u. Regarding claim 29, Yedid also discloses apparatus wherein:

- the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples (Column 4, lines 37-58. The symbol values imply that the samples have an average value. The DC reference value i.e. the claimed parameter is employed as a pseudo-symbol, implying that the fixed value of the parameter is substantially equal to the average value of the symbols i.e. the claimed samples).

5v. Regarding claim 30, Yedid also discloses apparatus wherein:

- the coefficient generator is further configured to receive a plurality of locally generated samples, and adapt the filter coefficient as a function of the locally generated samples. (samples in delay line 101 of Fig. 6 are generated locally due to an analog to digital conversion by an analog to digital converter (ADC) such as block 33 of Fig. 1. In Fig.1, the ADC is followed by an echo canceller which uses the adaptive filter. In column 6, lines 44-53, equation 1 explains how the coefficients are adapted as a function of the delay line samples 101).

5w. Regarding claim 31, Yedid further discloses apparatus wherein:

- the coefficient generator is further configured to adapt the filter coefficients by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples (Column 2, lines 9-22, where the LMS algorithm is a minimum mean square error algorithm and Column 6, lines 43-53, wherein equation (1) is the update equation used in LMS algorithm).

5x. Regarding claim 40, Yedid discloses a transversal filter for minimizing non-linear distortion in signals comprising:

- means for adapting a plurality of filter coefficients(Block 107 in Fig. 6; coefficients c_i in Fig. 6); and means for filtering a plurality of samples by applying one of the filter coefficients to a parameter (provided by DC tap 116 in Fig. 6), applying each of the remaining filter coefficients to one of the samples and combining the parameter and the samples (Summer 115 in Fig. 6 combines the samples and parameter); wherein the adaptation of the filter coefficients is a function of the combined parameter and samples. (Fig. 1, Fig. 6 and column 8, lines 31-60).

5y. Regarding claim 41, Yedid further discloses apparatus wherein:

- the means for filtering the samples comprises means for multiplying said one of the filter coefficients with the parameter (the DC tap value is multiplies by a coefficient at multiplier 114 in Fig. 6), means for multiplying each of the remaining filter coefficients with its respective sample (inputs to summers 111-1 through 111-4 are the products of filter coefficients c_i and their respective samples), and means for summing the parameter and the samples.(Summer 115 in Fig. 6 combines the samples and parameter). (Fig. 6 and column 8, lines 31-60).

5z. Regarding claim 42, Yedid further discloses apparatus comprising:

- means for serially receiving the samples(Block 101 in Fig. 6) (Fig. 6 and column 8, lines 31-60).

5a1. Regarding claim 43, Yedid further discloses apparatus wherein:

- means for adapting the filter coefficients uses a least square algorithm. (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm. LMS is a least square algorithm).

5a2. Regarding claim 44, Yedid further discloses apparatus wherein:

- the least square algorithm comprises a least mean square (LMS) algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm).

5a3. Regarding claim 45, Yedid further discloses apparatus wherein:

- the parameter comprises a fixed value (Fig. 6 and column 8, lines 31 60, wherein the fixed reference value implies that the parameter has a fixed value).

5a4. Regarding claim 46, Yedid also discloses apparatus wherein:

- the samples comprise an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples. (Column 4, lines 37-58. The symbol values imply that the samples have an average value. The DC reference value i.e. the claimed parameter is employed as a pseudo-symbol, implying that the fixed value of the parameter is substantially equal to the average value of the symbols i.e. the claimed samples).

5a4. Regarding claim 47, Yedid also discloses apparatus wherein:

- the adaptation of the filter coefficients are further a function of the locally generated samples. (samples in delay line 101 of Fig. 6 are generated

locally due to an analog to digital conversion by an analog to digital converter (ADC) such as block 33 of Fig. 1. In Fig.1, the ADC is followed by an echo canceller which uses the adaptive filter. In column 6, lines 44-53, equation 1 explains how the coefficients are adapted as a function of the delay line samples 101).

5a4. Regarding claim 48, Yedid also discloses apparatus wherein:

- the adaptation of the filter coefficients are performed by applying a minimum mean square error algorithm to the filtered samples and the locally generated samples (Column 2, lines 9-22, where the LMS algorithm is a minimum mean square error algorithm and Column 6, lines 43-53, wherein equation (1) is the update equation used in LMS algorithm).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

6. Claim 7 and 19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Yedid in view of Ruelke (US 6459889), hereafter referred to as Ruelke.

6a. Regarding claim 7, Yedid discloses all the limitations claimed (See paragraph 5a above).

However, Yedid fails to disclose the details of the DC bias adjustment of the samples.

In the same filed of endeavor, however, Ruelke discloses a DC offset correction loop for a radio receiver with a method comprising:

- monitoring a DC bias of the samples, and reducing the DC bias if it exceeds a threshold. (Fig. 2; Column 6, lines 57-64. Since a correction is made to the baseband DC offset i.e. the claimed DC bias of the samples only when the DC offset exceeds a predetermined threshold, the reference implies that the DC offset (claimed DC bias) is monitored and reduced upon exceeding a threshold).

Thus, it would be obvious to a person of ordinary skill in the art to employ the DC offset correction method before equalization as taught by Ruelke because Ruelke's method is optimized to provide a correction of DC offsets in a receiver. This hardware methodology is complimented by equalizing algorithms implemented on a digital signal processor (Column 3, lines 31-53).

6b. Regarding claim 19, Yedid discloses all the limitations claimed (See paragraph 5i above).

However, Yedid fails to disclose the details of the DC bias adjustment of the samples.

In the same filed of endeavor, however, Ruelke discloses a DC offset correction loop for a radio receiver comprising:

- an outer loop configured (Block 224 in Fig. 2) to monitoring a DC bias of the samples generated by the analog-to-digital converter, and reducing the DC bias if it exceeds a threshold. (Fig. 2; Column 6, lines 57-64. Since a correction is made to the baseband DC offset i.e. the claimed DC bias of the samples only when the DC offset exceeds a predetermined threshold, the reference implies that the DC offset (claimed DC bias) is monitored and reduced upon exceeding a threshold).

Thus, it would be obvious to a person of ordinary skill in the art to employ the DC offset correction method before equalization as taught by Ruelke because Ruelke's method is optimized to provide a correction of DC offsets in a receiver. This hardware methodology is complimented by equalizing algorithms implemented on a digital signal processor (Column 3, lines 31-53).

7. Claims 8,9 and 20-22 are rejected under 35 U.S.C. 103(a) as being unpatentable over Yedid in view of Coker et al. (US 6625235 B1), hereafter referred to as Coker.

7a. Regarding claim 8, Yedid discloses all the limitations claimed (See paragraph 5a above).

However, Yedid fails to disclose the details of filtering the samples prior to equalization.

In the same filed of endeavor, however, Coker discloses an apparatus and method for noise-predictive maximum likelihood detection, comprising:

- notch filtering the samples. (Fig. 1, Fig. 7c and Column 8, lines 51-59. Notch filter 44c in Fig. 7c filters the samples coming from analog to digital converter 19 of Fig. 1. The notch filter 44c is followed by a digital equalizer 22 in Fig. 7c).

Thus, it would be obvious to a person of ordinary skill in the art to a notch filter before equalization as taught by Coker because the DC notch filter renders the further processing blocks immune to DC offset, thereby providing DC offset compensation. (Column 8, lines 51-59).

7b. Regarding claim 9, Yedid and Coker disclose all the limitations claimed (See paragraphs 5a and 7a above).

Coker further discloses a method wherein:

- the notch is substantially at DC. (Fig. 1, Fig. 7c and Column 8, lines 51-59. DC notch filter 44c in Fig. 7c filters the samples coming from analog to digital converter 19 of Fig. 1. The DC notch filter 44c is followed by a digital equalizer 22 in Fig. 7c).

Thus, it would be obvious to a person of ordinary skill in the art to a notch filter before equalization as taught by Coker because the DC notch filter renders the further processing blocks immune to DC offset, thereby providing DC offset compensation. (Column 8, lines 51-59).

7c. Regarding claim 20, Yedid discloses all the limitations claimed (See paragraph 5i above).

However, Yedid fails to disclose the details of filtering the samples prior to equalization.

In the same filed of endeavor, however, Coker discloses an apparatus and method for noise-predictive maximum likelihood detection, comprising:

- a notch filter configured to filter the samples. (Fig. 1, Fig. 7c and Column 8, lines 51-59. Notch filter 44c in Fig. 7c filters the samples coming from analog to digital converter 19 of Fig. 1. The notch filter 44c is followed by a digital equalizer 22 in Fig. 7c).

Thus, it would be obvious to a person of ordinary skill in the art to a notch filter before equalization as taught by Coker because the DC notch filter

renders the further processing blocks immune to DC offset, thereby providing DC offset compensation. (Column 8, lines 51-59).

7d. Regarding claim 21, Yedid discloses all the limitations claimed (See paragraph 5i above).

However, Yedid fails to disclose the details of filtering the samples prior to equalization.

In the same filed of endeavor, however, Coker discloses an apparatus and method for noise-predictive maximum likelihood detection, comprising:

- a notch filter configured to filter the samples. (Fig. 1, Fig. 7c and Column 8, lines 51-59. Notch filter 44c in Fig. 7c filters the samples coming from analog to digital converter 19 of Fig. 1. The notch filter 44c is followed by a digital equalizer 22 in Fig. 7c).

Thus, it would be obvious to a person of ordinary skill in the art to a notch filter before equalization as taught by Coker because the DC notch filter renders the further processing blocks immune to DC offset, thereby providing DC offset compensation. (Column 8, lines 51-59).

7e. Regarding claim 22, Yedid and Coker disclose all the limitations claimed (See paragraphs 5i and 7d above).

Coker further discloses a method wherein:

- the notch filter is further configured with a notch substantially at DC. (Fig. 1, Fig. 7c and Column 8, lines 51-59. DC notch filter 44c in Fig. 7c filters the samples coming from analog to digital converter 19 of Fig. 1. The DC notch filter 44c is followed by a digital equalizer 22 in Fig. 7c).

Thus, it would be obvious to a person of ordinary skill in the art to a notch filter before equalization as taught by Coker because the DC notch filter renders the further processing blocks immune to DC offset, thereby providing DC offset compensation. (Column 8, lines 51-59).

8. Claims 32-39 are rejected under 35 U.S.C. 103(a) as being unpatentable over Yedid.

8a. Regarding claim 32, official notice is taken regarding the limitation in the preamble:

- Computer-readable media embodying a program of instructions executable by a computer program to perform a method of adapting filter coefficients. (It is obvious that any signal processing algorithms, for example, an algorithm or program of instructions to adapt filter coefficients, can be stored on computer-readable media and are executable by a computer program)[1].

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[1] To mention a few references:
Asano (US 6212144 B1)
Behrens (US 5754353 A)
Satoh (US 5818655 A)

Yedid discloses a transversal filter for minimizing non-linear distortion in signals with a method comprising:

- adapting a plurality of filter coefficients(coefficients c_i in Fig. 6); and filtering a plurality of samples by applying one of the filter coefficients to a parameter (provided by DC tap 116 in Fig. 6), applying each remaining filter coefficient to one of the samples, and combining the parameter and the samples(Summer 115 in Fig. 6 combines the samples and parameter); wherein the adaptation of the filter coefficients is a function of the combined parameter and samples.(Fig. 6 and column 8, lines 31-60).

8b. Regarding claim 33, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the filtering of samples multiplying one of the filter coefficients with the parameter (the DC tap value is multiplies by a coefficient at multiplier 114 in Fig. 6), multiplying each of the remaining filter coefficients with its respective sample (inputs to summers 111-1 through 111-4 are the products of filter coefficients c_i and their respective samples), and summing the parameter and the samples (Summer 115 in Fig. 6

combines the samples and parameter). (Fig. 6 and column 8, lines 31-60).

8c. Regarding claim 34, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the adaptation of the filter coefficients comprises using a least square algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm. LMS is a least square algorithm).

8d. Regarding claim 35, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the least square algorithm comprises a least mean square (LMS) algorithm (Column 2, lines 9-22 and Column 6, lines 43-53, wherein equation (1) is the update equation used in Least Mean Square (LMS) algorithm).

8e. Regarding claim 36, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the parameter comprises a fixed value (Fig. 6 and column 8, lines 31-60, wherein the fixed reference value implies that the parameter has a fixed value).

8f. Regarding claim 37, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the samples have an average value, and wherein the fixed value of the parameter is substantially equal to the average value of the samples (Column 4, lines 37-58. The symbol values imply that the samples have an average value. The DC reference value i.e. the claimed parameter is employed as a pseudo-symbol, implying that the fixed value of the parameter is substantially equal to the average value of the symbols i.e. the claimed samples).

8g. Regarding claim 38, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid also discloses a method wherein:

- the adaptation of the filter coefficients is further a function of a plurality of locally generated samples (samples in delay line 101 of Fig. 6 are generated locally due to an analog to digital conversion by an analog to digital converter (ADC) such as block 33 of Fig. 1. In Fig.1, the ADC

is followed by an echo canceller which uses the adaptive filter. In column 6, lines 44-53, equation 1 explains how the coefficients are adapted as a function of the delay line samples 101.).

8h. Regarding claim 39, Yedid discloses all the claimed limitations (See paragraph 8a).

Yedid further discloses a method wherein:

- the adaptation of the filter coefficients further comprises applying a minimum mean square error algorithm to the filtered samples and the locally generated samples (Column 2, lines 9-22, where the LMS algorithm is a minimum mean square error algorithm and Column 6, lines 43-53, wherein equation (1) is the update equation used in LMS algorithm).

Conclusion

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

- Beaudin et al. (US 6853694) disclose aspatial diversity wireless communications receiver.
- Lee (US 5345472) discloses a method and apparatus for receiving and decoding communication signals in a receiver.

Contact Information

10. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Vineeta S. Panwalkar whose telephone number is 571-272-8561. The examiner can normally be reached on M-F 8:30-5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Mohammad Ghayour can be reached on 571-272-3021. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

V.P.

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